Amendments to the Specification

Please delete paragraph **[0007]**.

Please replace paragraphs **[0006]**, **[0008]**-**[0016]**, **[0018]**-**[0020]**, **[0022]**, **[0023]**, **[0026]**, and **[0030]** with the following paragraphs set forth below:

[0006] Given this prior art, one of the objects of the ~~present invention~~ <u>various embodiments described herein</u> is to create a low-frequency circuit of the type referred to above that provides a high degree of adjustability combined with a simple, economical design.

[0008] In order to achieve high adjustability with a headroom that is typically increased by 6 dB, the ~~invention~~ <u>circuit</u> provides what amounts to dynamic compression, which in contrast to the previously used signal compression, is essentially limited to the audio signal peaks. Specifically, this compression is achieved by means of (two) nonlinear transfer elements that act on the audio signal in such a way that one of its half waves is processed in a nonlinear manner. Such half-wave-specific signal processing is performed for the non-inverted and the inverted audio signals which are then subjected to a subtraction operation so that the original audio signal is essentially compressed only in the area of its signal peaks. One of the main advantages of this amplitude compression, which is limited to the signal peaks of the audio signal, as compared to conventional control amplifiers, is that the claimed compression that takes place is free from slowness and audible control processes, as well as control intermodulations.

[0009] The amplitude compression ~~of the invention~~, which is limited to the signal peaks of the audio signal, is associated with the generation of a relatively low harmonics content, which the human ear does not perceive as distortion, but rather as a 'vitalization" of the signal, since the audio signal is not cut off at a fixed amplitude, but rather has its volume reduced depending on how the nonlinear performance curve of the nonlinear element is set up. An advantage here is that the audio signal retains its original characteristic, but on average has a louder and fresher impact as a result of the harmonics that are generated in the compression process.

[0010] The dynamic compression ~~of the invention~~ is extremely effective, mainly at capturing the signal peaks from electric instruments, such as electric guitars or electric basses, so that the average signal level and thus the cleanness of the audio signal is improved significantly.

[0011] Another important advantage ~~of the invention~~ is that the processing of the signal in later signal processing stages encounters fewer problems due to the reduced dynamic range, which results in better use of the resources of equipment employing the circuit ~~of the invention~~.

[0012] The low-frequency circuit ~~of the invention~~ ensures that the risk of overdriving is much lower than in prior-art circuits of the type being addressed here, which increases the subjective performance of equipment using the circuit of the invention, since these performance limits are not reached until much later.

[0013] The components used in the circuit ~~of the invention~~ are extremely simple, economical and trouble-free. For example, the nonlinear curves of the input stages of the circuit of the invention can, in the simplest case, be produced by diodes in passive input stages. In the case of passive input stages, amplification is provided in subsequent amplifier stages. Alternatively, other simple nonlinear transfer elements, such as bipolar transistors, FETs, vacuum tubes, etc., may be used in active -- i.e., amplifying - input stages.

[0014] If the audio signal is to be fed into the circuit ~~of the invention~~ in symmetrical form, a special circuit to generate an inverse audio signal is not necessary. In any event, an inverter of this type can be implemented at low-cost and only results in an insignificant increase in cost of the circuit.

[0015] Additional reduction of the dynamic range can be achieved by cascading two or more of the low-frequency circuits ~~of the invention~~.

[0016] Another advantage ~~of the invention~~ involves common mode suppression in the difference amplifier downstream from the input stages. Such suppression is needed to effectively reduce interference acting on the transmission path. The advantage relates to out-

of-phase transmission paths that are combined with one another and is especially important when vacuum tubes circuits are used.

[0018] The low-frequency circuit ~~of the invention~~ for the first time permits dynamic and harmonics processing directly in the input stage and thus provides processing operations in a compact circuit, operations that in the past had to be implemented in the number of different circuits.

[0019] We shall now illustrate ~~the invention using the~~ <u>selected embodiments with reference to</u> the <u>following</u> drawings ~~as an example of a preferred embodiment~~.

[0020] Fig. 1 is a schematic diagram of the general design of ~~the~~ <u>a</u> low-frequency circuit ~~of the invention~~ <u>according to one embodiment</u>

[0022] Figs. 3 to 7 represent signal shapes in the form of amplitude/time diagrams to elucidate the circuit of ~~the invention~~ <u>Fig. 1</u>.

[0023] Fig. 1 is a schematic diagram that shows the general design of a low-frequency circuit used to process audio signals. An audio signal is supplied by a signal source 10. In the embodiment shown here, this audio signal from signal source 10 has an asymmetrical shape. This means that relative to ground, the signal source 10 produces an audio signal having a specified phase orientation. This audio signal is supplied to the low-frequency circuit ~~of the invention~~. This circuit comprises a first input stage 11 and a second input stage 12. The two input stages, 11 and 12, preferably are designed in an essentially identical way and essentially have the same nonlinear performance curves. These input stages can be designed as passive elements, as will be described below based on Fig. 2. Alternatively, input stages 11 and 12 can be active elements based on amplifiers, etc.

[0026] Fig. 2 shows a specific embodiment of the circuit of Fig. 1. In cases when the same components are shown, those in Fig. 2 are indicated by the same reference numbers as those in Fig. 1. The nonlinear curve of the two input stages, 11 and 12, is provided in Fig. 2 by nonlinear elements in the form of diodes 15 and 16. Diodes 15 and 16 are grounded at their cathodes. A resistor 17 is attached ahead of the anode of diode [[12]] <u>15</u>, which belongs to the first input stage, while a resistor 18 is attached ahead of the anode of diode 16, which

belongs to the second input stage 12. The points at which the resistors are connected to the diodes each constitute the output of input stages 11 and 12, while the other resistor connections, 17, 18, constitute the input of input stages 11, 12.

[0030] Fig. 5 shows the output signal from difference amplifier 14, at whose two inputs signals A and B from Fig. 4 are applied. The resulting difference signal is identified in Fig. 5 by the letter C, and for comparison purposes, the non-processed audio signal is identified by letter D, which is supplied at the difference amplifier 14 by bypassing input stages 11 and 12. If one compares the curves of signals C and B, it becomes apparent that the audio signal that is partially processed in a nonlinear manner in input stages 17 and 18 is compressed in its peak range at the output of difference amplifier [[5]] 14. This means that the original sinusoidal curve is changed into a flattened sinusoidal curve through the processing that occurs in input stages 11 and 12. The difference between signals C and D in the range of their maxima and minima corresponds to an amplitude difference x, which represents the increase in headroom resulting from the processed signal C compared to the unprocessed signal D.